5.2 Release (09/20)

End user features 5.2 Release (09/20)

OpenSIP specific

Server Independent Intercom Auto-Answer: The auto-answer feature improves the end user experience when using the intercom feature, enabling VVX phones to act as true intercom devices where no action is required on the part of the called party to answer a call.

Benefit - Auto-Answer provides an improved user experience and improves end user productivity, particularly in Boss Admin environments. This feature enables the boss and admin (or any other phone to act as a traditional intercom system. The traditional intercom, where one will press a button, speaks into the intercom and the person on the other end hears audio without having to do anything. Up until 5.2.0, the person on the other end actually had to "answer" the call which means the boss is left waiting for a pickup before speaking.

Support for GENBAND E911 location tree: This feature allows the phone user to select their location from a preconfigured list of locations. This location is used in emergency calls and allows the GENBAND server to route the emergency call to the appropriate Public Safety Access Point (PSAP).

Benefit - The GENBAND E911 location tree allows the emergency services to provide a faster response time that provides ease of mind to the end user.

Improve VVX -GENBAND Interoperability by adding Private and Global Address Book integration: This feature adds support to the VVX line for GAB/PAB (Global Address Book / Personal Address Book).

Benefit - The GENBAND GAB/PAB support provides an improved end user experience allowing synchronization of address book information across phones and soft clients.

GENBAND MADN-SCA: The MADN-SCA (Multiple Appearance Directory Numbers – Single Call Arrangement) allows one telephone number that can be directed to multiple devices.

Benefit - It provides call state information to all members of a group of users who need to monitor the same line, including ability to pick up held calls and to "barge in" to active calls.

UCS - VVX General UI improvements:

Benefit - VVX General UI improvements provide simpler and improved end user experience and improved productivity, such as ability to use BLF in larger hunt groups and overall UI performance improvements.

Enterprise features 5.2 Release (09/20)

IT Admin Specific Features

Reduced size of the sip.ld file by 25% for the VVX phones: The reduced size enables the IT admin or the Service Provider to reduce the network bandwidth required for build deployments and improve the overall ease of deployment.

Benefit - Optimize download provides an improved deployment experience and makes it easier for service providers to keep customers on the latest UCS build.

FQDN Support for H323: FQDN (Fully Qualified Domain Name) support will be provided for H323 calls to allow failover to work efficiently and keep the phone operational for H323 calls in the event of a primary server failing.

Benefits - Reduces down time for the service and simplifying maintenance for customers by allowing failover for H323 to work similarly to SIP calls.

Control over SIP subscription overlap timers: New configuration parameters have been added to allow service provider administrators to have more fine grained control over the various SIP subscription timers.

Benefits: Allows service providers to control how the phones on the network will subscribe to the call server for event notification. This allows optimization of the whole network and reduce the load on large core networks. We can provide a list of configuration parameters with different values

5.1.1 Release (06/14) – Lync features

End user features 5.1.1 Release (06/14)

Shared Line Appearance (Boss/Admin) – Lync (support for Lync Boss Admin on VVX EM): This feature is designed for an Admin Assistant managing the phone of their manager remotely, using their own Polycom phone. With this release, the VVX expansion modules are supporting the Lync boss admin feature. The Shared Line Appearance (Boss/Admin) feature allows VVX phones configured as an “Admin role” for a “Boss” to redirect or transfer an incoming call for the “Boss” on the shared line key - e.g.: call can be sent to Boss’s voicemail.
Benefits: - The Boss Admin feature on the VVX EM improves admin users’ experience and improving their productivity.

Enhanced presence support on VVX phones: Enhanced presence extends the available list of presence states in a Lync environment. The newly supported presence states are: “In a Meeting”, “In a Conference”, “Presenting” and “Urgent Interruptions Only”.
These functions now work on the VVX phone and the VVX Expansion Modules (that are attached to the phone).
Benefits: - The enhanced presence support improves the visibility of boss or peers status and ultimately prevents interruptions and wasted calls thereby enhancing workflow in an organization.

Simplified Caller ID for Lync: The display indicates only what the end user needs to see in an improved and simplified view.
Benefits: - This feature provides a simplified view of caller details.

Ability to dial, transfer any number or to the voicemail accessible in the user’s Contact Card: The Contact Card feature will enable the end user to access any numbers that are presented in the Lync contact list and can be viewed and dialed directly from the phone.
Benefits: - Easy access in the Lync contact list.

Contextual Soft key to Redirect /Transfer call to Boss’ Voicemail: This feature enables VVX end points to be configured and used via the shared line key. The contextual soft key is accessible at the time of an incoming call or for an answered call.
Benefits: - This Shared Line Appearance (Boss/Admin) feature improves the admin user experience and efficiency.

Integration of the VVX EM for Lync/ provide Smart Paging for improved EM efficiency: When enabled, smart paging will automatically populate and fill up the VVX EM pages while making use of the visible VVX EM screen before moving on to populate the second and third pages of an LCD EM
Benefits: - This smart paging VVX EM feature improves the admin user experience and efficiency.

Enterprise features 5.1.1 Release (06/14)

Lync Root Certificate Retrieval Using LDAP: This feature allows the Polycom VVX portfolio of products to register in a Lync environment without the Administrator’s intervention of installing the root certificate chain when the specific DHCP option 43 is NOT provisioned. This feature allows VVX phones to automatically obtain the Root certificate or root certificate chain, using LDAP to provide an Out Of the Box Experience (OOBE) to the Lync user.

Benefits: - Improves installation efficiency and reduces the overall installation costs.

PIN Authentication through Web UI: PIN authentication via the web configuration utility enables the end user to have an easy way to update the pin on the SoundStructure product which is often rack mounted and does not have an easily accessible "dial pad"

Improved Security – Secure by Default: If an IT Admin enables the web server by default, the web UI is only accessible via HTTPS to ensure security. The HTTP access can be enabled by the IT Admin.
Benefits: - Improves the security of the phone by preventing unauthorized users to directly access the phone data through the Web UI. This new security feature provides true improved security.

Visual indication of Security Classification of calls on BroadSoft: This feature allows the Polycom VVX portfolio of products that are deployed via the BroadSoft Call Control platform and send a visual security indication to the end point. The feature enables an end user to have a security classification associated with their lines.
Benefits: - Improves the security and visual indication of calls however still preventing unauthorized users to directly access the phone data. UI. This new security feature provides true improved security.

BroadSoft server based call recording: This features enables server side functionality such as enabling call recording and storage on the server.
Benefits: - Improves the security and call recording of the phone by preventing unauthorized users to directly access the phone data. UI. This new security feature provides true improved security.

On hook protection – Disabling the hands free speaker and microphone: This features enables on hook protection which disables the hands free, speaker and microphone if the phone is on hook.
Benefits: - Improves the security of the phone. UI. This new security feature provides true improved security.

Lync Data Center Resiliency (also known as Front End Pool Pairing):
During an outage of a Lync Server Data Center or any disconnection of the Front End Server all the users registered to that registrar pool move to their secondary registrar. After the primary Front End server recovers the users move back to the original data center.

- Outages due to connectivity or the backend server failures are supported with notifications to the endpoint.
- Support migration between registrars signaled via NOTIFY requests.

Customers want the same "5 nines" reliability that PBXs with failover resources deliver.
Benefits: - This feature makes achieving "5 nines" reliability possible.

5.0 Release (08/13) – Lync End user features, OpenSIP End user features, Admin Specific features

End user features – Lync specific

Better Together over Ethernet: BToE (Ph-1) – Lync - This feature is intended to provide the user an enhanced experience when the Lync Compatible IP Phone is working directly with the Lync Client using Ethernet tethering. The PC running the Lync client is connected to the PC port of the phone and LAN port of the phone is connected to the network.

The Polycom phones are envisioned to be tethered to the PC using the Ethernet interface. The phones are supposed to work in one of two modes:

1. **Paired Mode**: Phone emulates itself as both virtual USB HID device and virtual USB audio device
2. **Audio Playback Mode**: Phone emulating itself as a virtual USB audio device.

Benefits: -Enhanced end user experience from phone to Lync Client using Ethernet tethering.

Shared Line Appearance (Boss/Admin) – Lync: The Shared Line Appearance (Boss/Admin) feature is designed to add greater flexibility to the user experience when using Polycom VVX phones with Microsoft Lync. This feature is designed to support an Admin Assistant managing the phone of their manager remotely using their own Polycom phone.
Benefits: - The Boss Admin feature improves admin users' experience and improving their productivity.

Polycom UC Software Feature Summary

Call Park – Lync: The Call Park feature allows the user to place a call on hold, so that it can be retrieved from another phone (for example from the phone in a conference room). If the user is on an active call, the user can park the call to a call park extension by pressing the Park soft key or the Call Park button. Any user is able to dial the Call Park extension to retrieve the call
Benefits: - Simple way to park a call or retrieve the call.

Address Book Service: ABS – Lync: Address Book Service the feature that provides the ability for Polycom phones to search through the Microsoft address book.
Benefits: - Simple search of address book.

Lync 2013 interoperability: With this release we have achieved the Microsoft Unified Communications Open Interoperability Program (UCOIP) for a list of Polycom telephony devices.
Benefits: -true Lync interoperability

End user features – OpenSIP specific

Verify Missed Call tracking on non-shared lines – OpenSIP: -
Phone not beep/light up if MWI (OpenSIP.Message Waiting Indicator) notify received with no change in MWI status.
Benefits: Easy call tracking capability.

Admin Specific Features

Polycom® BroadSoft UC-One Integration (new feature on the VVX 300-400)
Support for BroadWorks Presence Capability (XMPP) —Allows users to share presence information with the BroadTouch Business Communicator (BTBC) client application
Benefits:- Enables users to share presence details.

Support for BroadWorks Directory Integration (access directory on the server and your personal directory) —Displays information for all users in the enterprise; for example, work and mobile phone numbers. End user can have an integrated toolbar that enables them to speed dial and use the directory.
Benefits:- Integrated toolbar and easy access to the directory.

Support for BroadSoft favorites —Allows users to contacts marked as favorites with the BroadTouch Business Communicator (BTBC) client application. Favorites enables the end user to build a list of people they contact most often.
Benefits:-Enables easy listing of most often used contacts as favorites.
Lync in-band device update - Lync: - The provisioning and update of software for VVX® phones are managed through the Lync server. Lync in-band device update is an automated straightforward process to update to the latest version and it's provisioning. This requires the normal Polycom ® Software package to be repackaged in a format compatible with the Lync server software management application. This feature includes the following items:

1. Initial device provisioning.
2. Moves/Changes/Adds/Deletes (MACD) to assign /re-assign phones to users.
3. Automated software updates
4. Quality of Service Monitoring.
5. Logging and error reporting.

Benefits:-easy provisioning and software update via Lync in-band device update process.

Synchronized (ACD) Automatic Call Distribution (Enhanced & Premium): - This feature ports the existing ACD functionality from the SoundPoint IP® phones to VVX® phones. This includes the ability for phones to work in a BroadSoft based call center and to receive calls based on agent availability, workload etc.
Benefits:-provides synchronized call distribution.

Premium ACD: Port Hoteling and Call Center Status Threshold: - VVX® phones are capable of supporting the BroadSoft hotelling feature, allowing guest users to sign in to a phone. Once signed in, the phone registers with the server to receive Call Center Events and populates call queues associated with the guest. The Status Threshold feature ensures that the various call queue thresholds for each call center an agent is monitoring are displayed on the VVX ® display.

The UC 5.0.0 software release supports the VVX 3x0, 4x0, 500, 600 and 1500 products.
This release is introducing a set of new VoIP end user and admin features:

- ***Lync Telephony** functionality required for large scale adoption of Polycom VVX products for use with a Lync telephony solution. A set of features required to improve the usability and manageability of the VVX products for all '**OpenSIP**' partners to enable existing Polycom customers to migrate their phone offerings from SoundPoint IP to VVX products.*

Benefits:-enables guest users to sign in via any phone.

4.1.3RevG Release (06/12) – BroadSoft UC One features Release

Polycom® BroadSoft UC-One Integration

The Polycom BroadSoft UC-One application integrates with a BroadSoft enterprise directory and BroadCloud services—a set of hosted services that BroadSoft runs—to provide three features on Polycom® VVX® 500 and VVX 600 business media phones:

Broadsoft Directory—Displays information for all users in the enterprise; for example, work and mobile phone numbers
Broadcloud Presence—Allows users to share presence information with the BroadTouch Business Communicator (BTBC) client application
Directory Integration— Search and display of BroadSoft enterprise directory contacts; for example, work and mobile phone numbers
UC One Presence— Unified presence information with the BroadTouch Business Communicator (BTBC) client applications on other devices
UC One Contacts and Favorites— Synchronized UC One Contacts, Personal Groups and Favorites with BroadTouch Business Communicator (BTBC) client applications
These features require the BroadSoft BroadWorks R18 SP1 platform with patches and the BroadSoft BroadCloud services.

UC Software 4.1.3 Rev G supports the following Polycom endpoints:

- *VVX® 500 business media phone*
- *VVX® 600 business media phone*
- *SoundStructure®*

Benefits:-A set of UC-One integration capabilities enabling directory, presence, and favorites to be used via UC One client.

4.1.3 Release (02/13) – VVX Camera Release

Added support for VVX camera– Attaching the VVX Camera via USB to your VVX 500 or VVX 600 converts your desktop phone to a video endpoint.
Benefits – Allows users to use Video communication right from their desk.

SIP and H.323 video codec – VVX Camera Supports both SIP and H.323 protocols for video communication on VVX 500 & VVX 600.
Benefits – Provides a variety of codec choices for communication according to the network availability and requirements.

Note: UC software 4.1.3 does not support Lync video on Microsoft Lync Server.

4.1.2 Release (12/12)- VVX600 Release

Added Support for Polycom VVX600 Product

Pin Authentication – Allows the user to use phone-number/Extension and PIN for the authentication instead of entering all the user credentials (Domain, Username, Password etc.,) to obtain the web ticket and sign in to the Lync server.
Benefits – Provides a simple solution for quick sign in.

BroadSoft Hoteling Event package – An interface defined to allow the phone to synchronize with the Hoteling guest service. It also provides an interface for creating and terminating Hoteling guest-host associations using message digest authentication on VVX 500 and VVX 600 phones.
Benefits - The call center agent sits at a Hoteling workstation and uses the SIP phone to sign in as a guest on the workstation, change their ACD state and start handling ACD calls.

4.1.0B Release (10/12) - Lync Qualified

Microsoft® Lync® Server 2010 Integration
Microsoft® Lync® Server 2010 is a unified communications (UC) solution that enables customers, colleagues, and business partners to communicate instantly by voice, video, or messaging through a single interface, regardless of their location or network. Administrators can configure Microsoft Lync to work with their existing call servers and also configure Lync directly into existing infrastructure, eliminating the need for additional gateways. This allows users to use Lync directly from the phone's user interface.

*Native OPEN SIP features are available, but they have not been fully validated. This release is intended for use in Microsoft Lync environments.

Lync Call and Local Features

Call Admission Control (CAC)
Prevents oversubscription of priority queue, an administrator is provided with a mechanism to throttle bandwidth usage for Lync voice and video calls (For Lync only).
Benefits - Allows administrators to control network traffic more efficiently by protecting network bandwidth from Lync Server traffic consumption, thereby maintaining high quality calls for all users in the network.

Call Detail Records (CDR) – Reports on calls, inventory of endpoints, and media information.
Benefits - An effective means for troubleshooting, inventory management and call cost accounting and verification.

Call Forwarding – The phone allows all calls to the user's phone to another Microsoft Lync contact or to voicemail.
Benefits – Provides a simple solution to forward any calls to another destination.

Contact Groups – Displays and expands Groups in the Lync user's contact list.
Benefits – Find any contact quickly by organizing multiple contacts into various categories.

Contact List and Presence – Allows the user to configure contact list from Microsoft Lync contacts. The Frequent Contacts feature creates a cache to keep track of communication patterns to automatically add contacts. Selected contacts can be monitored in real time. The contact card allows the user to quickly look up contact details and start an IM, call, or email session.
Benefits - The user can quickly find and communicate with any Lync user.

Call logs (missed, dialed, all, etc.) – Contains call information such as remote party identification, time and date, and call duration in three separate lists; missed calls, received calls, and placed calls.
Benefits - Allows the user to manage missed calls easily and quickly, preventing lost opportunities and business.

Peer-to-Peer Audio Calling – Initiate and receive two-party calls from any supported Lync endpoint.
Benefits – Convenient method for users to communicate with remote Lync users.

Private Lines - Alternate call-forwarding identity for a Lync user's secondary DID.
Benefits – This number can be used as an alternate exclusively number known to selected people. Best use case would be for Executives.

Lync Network and Security Features

Federation - Extends the Lync Server 2010 capabilities over the Internet for users to communicate with other organizations and companies that run Lync Server 2010 or previous versions.

Benefits – Users have a variety of ways to communicate with other end users in different companies and organizations and also view their statuses. It helps reduce costs of public phone network usage as Federation is conducted over the Internet.

Monitoring (Device Inventory Reports) - Records all communication activities and creates reports in four categories: System Usage, Call Diagnostics, Call Diagnostics for every user, and Media Quality Diagnostics. *Benefits – Facilitates identification of any particular issue and provide information for all communication sessions in the company or organization.*

Team-Call – Allows a group of contacts that can answer incoming calls to any member of the group. *Benefits – Maintains fast call pick-up time as any user within the same configured group can answer for another user who is away from the desk.*

Branch Office Survivability Reporting – Maintain SBA/SBS registration during WAN outage, automatic recovery.
Benefits – Branch offices are able to maintain network connection to each other during network connectivity failures. Minimizes down time, maintains productivity, and prevents loss of revenue.

E911 – Support of in-band provisioning information for Emergency 911.
Benefits – Ensures emergency calling with location information, reducing the need to communicate location information to the dispatcher.

Media Bypass – Support for by-passing the Lync mediation server to send media directly to PSTN Gateway. *Benefits –Enhances media quality and transmission, reduces WAN bandwidth, and utilizes G.711 codec to improve audio quality.*

Delegates – Other users can be delegated to send and receive calls on behalf of another user.
Benefits – Reduces missed calls and increases productivity as workers are kept up-to-date with important tasks.

Remote Worker (Edge Registration) – Supports endpoints outside of organization with secure media encryption.
Benefits –Allows flexibility and portability for users to work beyond the office space without the need for additional equipment.

WebTicket Service Compatibility – TLS-DSK Authentication.
Benefits – Provides for native Lync authentication used when NTLM is not available (e.g. Failover scenarios).

Additional Lync Network Capabilities
Dial Plans - Supports Lync Server Regex normalization patterns passed via in-band provisioning to the endpoint. Limited to regular expression support. Option to do server side normalization.

Provisioning – Support of in-band provisioning from Lync Server. Transmit configuration information to Lync client applications through SIP.

VLAN Assignment – LLDP –MED VLAN assignment. Enables discovery of LAN policies, device location discovery, extended and automated management of PoE end points, and inventory management.

Additional Lync Features
Response Groups – Routes calls within the system to the correct agents based on configured rules.
Benefits – Support for both calling response groups and receiving calls as a member of response groups.

Device Sign-In – Out-of-the-box device user sign-in and sign-out with cached credentials.
Benefits – Allows users to sign in to their own devices using the same set of Active Directory credentials they use on their workstations.

Root Certificate Download – DHCP Option 43
Benefits – Automatic provisioning of internal root CA certificates to devices.

TCP Media – RTP media and ICE negotiation supported over TCP when UDP is unavailable.
Benefits – Provides fall-back path media in scenarios where UDP communications fail.

Audio Codec Support – G.711, G.722-1 (All SoundPoint IP, VVX, SL 84xx, SSIP 5000, and SS Duo products).

Native SIP Features
Dial Plan Normalization – Allows the administrator to modify dialing configurations.
Benefits - Centrally manage dialing rules for users.

Message Waiting Indicator (MWI) – All phones alert the user to incoming text messages visually on a physical MWI lamp on equipped devices or aurally with audio alerts established.
Benefits – Provides a quick and familiar way to communicate instantly.

Network, Provisioning, and Server
Presence Status & Control – Presence settings can be changed on the phone to indicate the user's status to other Lync contacts. A status change on the phone also affects the status on the Microsoft Lync client. A menu of presence status control is available for users to manually select their presence icons.
Benefits – Allows full control of the user's work environment and indicates the status of the worker in real-time.

Log Access – Local access to device diagnostic logic.
Benefits – Used for support and troubleshooting activities.

Device Updates – Centralized device update from out-of-band server.
Benefits – The phone checks for new updates and updates the firmware automatically.

4.0.3 – 4.0.1 Release

UCS 4.0.3 Rev F (10/12) Features – Limited Release
Added Support
telURi in the P-Preferred-Identity header of INVITE messages
Only RFC 3264 type SDP media negotiation allowed

UCS 4.0.2B (04/12) Features
Added Support
DHCP renew after loss and recovery of WiFi LAN connection (SpectraLink 8400)
Enhanced digitmap by Removal of prepending "+" to outbound calls and giving the option of configuring the "+" in the dial plan (Applies to Lync mode only)
Early media followed by local ring back

UCS 4.0.2 – Limited Release
Added Support
XT9 PinYin input for Chinese Characters (VVX 1500)
BroadSoft Hoteling Event Package
BroadSoft Call Center Status Event Package
SpectraLink 8452 Wi-Fi handset with 2D barcode reader

UCS 4.0.1 (12/11)
Call Features
Unified Call Appearance List – Integrated into the existing phone behavior. Displays all calls on every line collectively in a single list. Users can scroll through the unified call appearance list to see which calls are on which line keys.
Benefits - Simplifies call management and usability when handling multiple calls. No need to change any configuration parameters.

Improved Call List Management - Call lists are non-volatile and persist after a phone reboot (VVX 500/1500 and SL8440 only). Calls answered on one phone are not logged as missed calls on other phones when using shared lines (for BroadSoft and Sylantro call servers).
Benefits - Easily organize and manage call-related tasks.

Instant Messaging (SpectraLink 8440/50/52 only) – The user can send and receive instant text messages with the handsets. A

Polycom UC Software Feature Summary

Message Waiting Indicator (MWI) LED alerts the user to incoming text messages visually on a MWI lamp on equipped devices or aurally with audio alerts established.
Benefits – Provides a quick and familiar way to communicate instantly.

Flexible Call Appearances – Organize registrations, line keys per registration, and concurrent calls per line key.
Benefits - Quick and easy way to manage and place calls.

Voice Mail Retrieval – Voicemail is saved at a centralized location for the user to access directly from the phone. New voicemail messages are indicated by a Message Waiting Indicator and a visual notification of the number of new voicemails on the phone's screen.
Benefits – One-touch call to voice mail attendant.

Merge Active Calls – Combine multiple active calls to initiate or add to conference calls.
Benefits – Saves time and enhances collaboration without restarting calls.

Hold, Mute, Transfer – Basic features for day-to-day communication.
Benefits – Familiar calling features allows users to be productive without a learning curve.

Local Features
Faster Boot Time - Reduction of time between phone reboot and obtaining a dial tone.
Benefits - Users have a quick start to the work day and make important calls on time.

Improved Boot-Up Behavior - Phone boots-up without a delay even if it lacks a network connection as a result of a configuration error.
Benefits - Desktop phone is ready to use whenever it is needed.

Reset Phone To Factory Defaults - The user can use the phone to clear overrides, speed dials, boot menu parameters, and phone data while maintaining the current software version.
Benefits - Reset your phone to factory settings without the need to send it back to the factory.

Predictive Dialing - "Smartphone" like predictive dialing for both text and numbers through call history and contact directory features for VVX 500 and SpectraLink 8400 (call history only for VVX 1500).
Benefits - Fast and easy dialing (especially with PDC), allowing enhanced user experience and efficiency.

PinYin Character Entry (VVX 1500) - Uses Nuance XT9® Smart Input to allow users to enter Chinese characters into text input fields using the phone's dial pad keys or through the onscreen keyboard.
Benefits - Commonly used text input method for Chinese characters.

Next-Gen Contact Directory - Includes "smartphone" directories (VVX 500/1500), unified call lists, configurable call list view, and option to edit the call entry before dialing.
Benefits - Familiar "smartphone" experience via flick-scroll, easy call list search, and contact entry customizations.

Flexible Home Screen Layout (VVX 1500) - Displays up to 29 line appearances, direct station selection (with presence) and speed dials (VVX 1500 only), used in conjunction with CMA. The VVX displays up to an additional 24 stations (with presence) or speed dial buttons and the "More" button enables full screen view if more than 5 soft-buttons are configured.
Benefits - Improved effectiveness for answering positions, increase handling of calls, and more efficient screen utilization.

Audio-Video Toggle (VVX 1500) - Provides options for users to toggle modes between video calls and voice- only calls.
Benefits - Enhances the user's experience by providing control over the medium of communication.

Multiple Language Support – Set on-screen language to your preference. Select from Chinese (Simplified and Traditional), Danish, Dutch, English (Canada, United Kingdom, and United States), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese (Brazilian), Russian, Slovenian, Spanish (International), and Swedish.
Benefits – Users have flexibility in working in their preferred language.

Polycom UC Software Feature Summary

Flexible Line Key Assignment - Allows users to assign line key attributes (line appearance, speed dial, Busy Lamp Field (BLF), presence) to any line key on the phone or expansion module. Enabling this feature overrides the existing automatic line key assignment behavior.
Benefits - Personalized user experience and flexibility for phone manageability.

Notification Profiles (SL 8440) - Users can customize alert tones for phone events such as incoming calls, receiving instant messages or pages, docking the handset, placing calls on hold, and low battery.

Benefits - Customizable alert tones enable users to personalize their handsets.

Interoperability and Integration

Polycom Desktop Connector - Allows sharing of keyboard and mouse between the user's computer and Polycom device (VVX 500/1500). Use the keyboard to enter text using PC keyboard (all languages are supported: German, Chinese, Arabic, etc.), use the mouse to easily navigate and select the phone menus (especially useful for VVX 1500 Web Browser), and copy text from the PC to phone menus.

Benefits - Polycom device becomes an extension of the user's desktop PC, as the PDC greatly improves user experience and efficiency.

Exchange Calendaring (VVX, SpectraLink 84xx products only) - Enables end user to view calendar items on Polycom desktop device. In order to use this feature the user has to sign-in, using their Exchange credentials, upon which they can start viewing their Calendar details.
Benefits - Increase productivity by allow the end user to click-and-dial into the appropriate meeting invite, without the hassle of memorizing bridge numbers and looking for details in Outlook.

Multicast Group Paging & PTT - The end user can send one-way audible pages to one of multiple available groups of phones. The audio page is played over the receiving phones' speaker.
Benefits - No need for overhead loud speakers. Subscription based peer-to-peer signaling. This feature is call server independent.

Network, Provisioning, and Server
Native Registration – The phone can register directly to the Lync Server without using a signaling gateway, sip proxy, or back-to-back user agent.

Enterprise Voice - Allows phones to connect with PCs, video devices, and other telephony endpoints in order to communicate via a combination of telephony features with presence, messaging, and conferencing capabilities.
Benefits – Identify the best method of communication with other users to increase productivity and enhance user experience.

Location Services – Send reports to Ekahau® Real-Time Location Systems (RTLS) on the SpectraLink handsets.
Benefits - Safe and efficient information delivery.

Web Configuration Utility & Provisioning - Enables easy configuration and provisioning of phone's features and user interface from a remote PC via multiple Web browsers.

Benefits - No need for technical knowledge, efficiently navigate through the phone's settings and features for greater productivity.

Distributed Polling for Software Upgrades - Phones automatically check their configuration files for new upgrades. Individual phones poll the provisioning server for upgrades at certain times specified in the phone settings.
Benefits - Customizable provisioning times can reduce server congestion.

Geographical Redundancy Enhancements - There are three behavioral enhancements. Re-registration on Fail Over requires the phone to be successfully registered with the RROFO-SBC before any communication takes place. Routing requires the phone to communicate with the server that processed the last successful transaction rather than always contacting the primary server. Detection identifies failed servers and prevents the phone from communicating with a failed server.
Benefits - Improved reliability for networks which use a RROFO-SBC network implementation.

Polycom UC Software Feature Summary

User Accessible Network Diagnostics - Ping to check if an IP connection can be established with a host. Also trace route between the phone and a far-end host.

Benefits - Users can troubleshoot network connectivity problems in their wired and wireless networks.

Warning and Error Notification Management - Improved methods of notifying user or errors and warnings, as well as improved management of warning messages.
Benefits - One menu for all warnings; faster resolution of problems.

Security
Improved Management of Trusted CA Pool - Create custom certificate profiles and choose which profiles are used for: 802.1x, Syslog, Provisioning, SIP, Presence, Browser, and LDAP. Increase in the number of supported customer certificates. Benefits - Improves security and convenient certificate management.

User Profiles - A unique profile which stores the user's personal phone information (contact directory, speed dials, phone settings) onto the provisioning server. Users may access their profiles anywhere using any phone connected to the provisioning server, and can also log in on more than one phone at a time.
Benefits - Provides greater flexibility for remote and mobile users and deployments which have common phones.

TLS Profiles - Configure the phone with a profile that specifies trusted digital certificates. Also allows the user to install and specify custom certificates.
Benefits - Conduct and control work-related communication securely.

Custom Device Certificates – Phones have a Polycom signed certificate installed at the factory, but are configurable to use either Polycom or Custom Certificate. They also support the ability to create a private or public key pair to generate a certificate.

Benefits - Improved security and flexibility for phone manageability.

BootROM (Updater) Enhancements - It seeks and installs new changes to the Updater software upon start/boot/reboot of the phone. Upon examination of the master configuration file, it downloads necessary application files, installs the applications into RAM, then uploads an event log file.
Benefits - Simplified application updates saves time and allows user to work worry-free from software changes throughout the day.

802.1x Authentication – Authentication methods include MD-5, EAP-PEAP, EAP-FAST, EAP-TLS, and EAP-TTLS.
Benefits – Validate and authenticate the client device that is attempting to connect to the LAN/WLAN.

SRTP encrypted media – Secure Real-Time Transport Protocol (SRTP) provides a way of encrypting audio streams to avoid interception and eavesdropping on phone calls.

Benefits – Phones negotiate the type of encryption and authentication to use for the session with the other end-point, which allows safe and secure connection for remote calling and federated communications conducted outside the company.

Audio Processing Features

Acoustic Echo Cancellation – Employs advanced acoustic echo cancellation for hands-free operation. Both linear and non-linear techniques are employed to aggressively reduce echo while providing natural full-duplex communication patterns.
Benefits – Facilitate a smooth and natural audio conferencing and reduce listener fatigue.

Audio Codec Support – G.711, G.722-1 (SoundPoint IP Products only).

Additional Capabilities
Addition of Host Name for DHCP Registration Ability to auto-answer an Incoming Call with a Muted Microphone Persistent Ringer volume after phone reboot Secure BootROM (Updater) HTTPS Provisioning USB Keyboard Support (VVX 1500)

Polycom UC Software Previous Version Features/Capabilities Summary

Polycom UC Software Feature Summary

SoundPoint® IP SIP is a feature-rich enterprise-class voice communications terminal for Ethernet TCP/IP networks. It is designed to facilitate high-quality audio and text message communications by a human user. It is an endpoint in the overall network topology designed to interoperate with other compatible equipment including application servers, media servers, internetworking gateways, voice bridges and other endpoints.

SIP is renamed as UC Software starting from Version 3.3.0. For detailed Release Notes and the SIP Downloads Matrix, please visit: http://support.polycom.com/PolycomService/support/us/support/voice/soundpoint_ip/previous_voip_softw are.html

UC Software (3.3.5 – 3.3.0)

UCS 3.3.5 Features (12/12)

Changed Polycom logo on the BootROM
Increased DNS-TTS cache parameter's maximum value from 65535(18 hours) to 2147483647 (68 Years)

UCS 3.3.4 Features (01/12)
Added Support
RFC 2782 full support

UCS 3.3.3 Features (11/11)
Added Support
Call Back – enabled using an Enhanced Feature Keys (applies to VVX 1500) Enhanced API for the VVX 1500 browser Webkit

UCS 3.3.2 Features (09/11)
Updates

Updated dial behavior – Added user confirmation on the phone before placing outgoing calls as part of the click to dial behavior.

Missed-Call Synchronization – When local call lists are disabled on the phones, Missed Calls notifications are sent from the call server to the respective users.

Display enhancements – Simplified display option by removing protocol tag & host details (VVX 1500D). Updated configuration files – Added functionality to set "NO Answer ring count" to "1" via menu options/configuration files (applies to VVX 1500D).

Zero Touch Provisioning (ZTP) feature support Geographical redundancy enhancements Support for Sennheiser EHS headset (phone menus and configuration) Extension of dialplan.digitmap String (supports up to 100 from 30 segments)

UCS 3.3.1 Features (11/10)
Updates

Feature Key Synchronization – Support added using FAC/NOTIFY message combination.
Benefits – Hosted IP solutions implement Synchronization of Feature key Functions(e.g. DND/CFWD) using a Feature Access Code (FAC) to set the Feature, and a SIP NOTIFY message to inform the phone of the feature state.

Team Function – Extends compatibility of statically configured Busy Lamp Field (BLF).
Benefits – Operates in a system requiring the use of two URIs (one for call operations and another to subscribe for notification of dialog events). Also provides Ringing Indication and a Directed Call Pick-Up capability in a system that does not generate RFC 4235 compliant dialog-info+xml documents.

Updated Phone Lock – Phones now display the full text of strings (applies to SoundPoint IP 320, 321, 330, 331, and 335).

Updated Idle Display – Phones now display the Call Forward destination.

Polycom UC Software Feature Summary

Added Support
Server certificate Serial Number (SN) Verification – Verified against the server/proxy's A record domain names if the SRV record domain does not match the SN.

Verification of authentication tag – Phones now provide a configurable parameter that allows the verification of authentication tag to be disabled for received SRTP packets.
Benefits – Allows system administrators to resolve defects in other endpoints where the authentication tag is not computed correctly. Supported parameter: sec.srtp.noAuthRxRTP.

LAN link recycling – During the 802.1x – EAPOL Logoff, the phone recycles the LAN link (e.g. it brings it down and up in an interval of one second) upon detecting a PC link down event.
Benefits – The 802.1X switch refreshes the authorized port state and starts to send request for identity challenge messages.

Bind authentication – Supported by Corporate Directory LDAP initialization. CMA presence (VVX 1500) Reboot/Configuration update from micro browser Premium extensions to server synchronized ACD feature Configurable toolbar slide-out option (VVX 1500)

<u>**UCS 3.3.0 Features (06/10)**</u>
Updates
Enhancement of visual indicator of incoming calls (VVX 1500 only)
Conference initiator configuration parameter - Control the behavior of terminating a 3-way conference. Terminate all conference legs or allow other parties to stay connected.

Slide bar (VVX 1500) - Adjust levels in various menu screens.

Timestamp - Displayed in Cal Lists alongside the Caller ID to keep track of call time.

API Telephony Event (XML) - Upon a successful line registration with a PBX, the API Telephony Event is sent to the attached application.
Increased maximum size of contact directory contact field (128) - Accommodates complex dialing scenarios.

Expanded configuration Web interface - Includes parameters associated with security.
Warble.wav file - This can be configured as an audible ringer for incoming calls, which generates a loud ringer tone for phones deployed in areas with a high ambient noise background.

Diagnostic menu option - Enables the display of configuration file statistics.
Updatable configuration parameter values at run-time Enhanced user interface selection for a distinctive ringtone associated with a contact in the local directory.

Self-contained configuring parameters (configuration process is more fault-tolerant) Expansion of the range of ports and randomization of port selection support for dynamic support of G.729AB and iLBC codecs (applies to SoundPoint IP 320, 321, 330, 331, 335, 450, 550, 560, 650, 670) Simplified codec configuration preferences

Lock and restriction of phone access from unauthorized users: Configuration parameter to obtain Caller ID from the From header (instead of the P-Asserted-Identity segment)

Local contact directory matching with Polycom CMA products style and user experience. Removal of redundant levels of abstraction associated with arrays in configuration files Optimization of RAM disk configuration parameters Minimal latency allowance to meet JITC requirements Caller ID information horizontal scrolling Configuration parameters support for TLS cipher suites Website usage of the new configuration system MD4 encryption key (OpenSSL) Enhanced Local Directory Search confirmation (SoundPoint IP 320, 321, 330, 331, and 335)

Expanded edit fields for additional content (VVX 1500) Presence and BLF support on Avaya CS2100 soft switches Activation and modification of registration parameters without phone restart or reboot requirement Automatic upgrade of BootBlock section of the

Polycom UC Software Feature Summary

BootROM On-board Web interface to obtain BootROM and application software versions Hide tool bar (VVX 1500) Ringtone: **Precedence call offered to phone rings with a corresponding precedence ringtone**: Ringtone: Initiation of precedence outgoing call generates a precedence style ring-back tone Aligned DSCP Differentiated Services Code Point levels (for standard and precedence level calls)

Display of current precedence level of a call Generation of MLPP resource-priority: Header based on the dialed number LogOut soft key display (enabling option on the SoundPoint IP 7000) Dynamic codec switching Enhanced computation of jitter buffer parameters (based on Quality of Service and expected payload size values) Increased default maximum call data rate (768 kbps from 512 on VVX 1500) Shortened user video call rate setting parameter value options (VVX 1500)

Enhanced browser rendering performance (VVX 1500)

Custom ring classes (se.rt): Can be set to a maximum value of 17

Custom ringer chords (tone.chord.ringer.spareX): Can be set to a maximum value of 19

Added Support
Diacritic letters and ligature support: ä, ö, ü / Ä, Ö, Ü ß - Greater language option selection. The diacritic letters and ligature can be displayed without having to change the character encoding scheme.

Custom device certificates installation capability - The administrator can add private and public keys (certificate) via TLS links.

Null Ciphers - Null Ciphers can be used with TLS Authentication. NTLM version 2 authentication [via XMPP, LDAP and HTTP(s)] for use with CMA G.719 audio codec in H.323 calls (VVX 1500) Asymmetric audio codecs Additional language support on the Advanced LDAP Search screen

SIP Application (3.2.7 - 1.0.9)

SIP 3.2.7 Features (06/12)

- Enhanced DNS TTL parameter (maximum value 2^32, or 2147483647 seconds)
- VeriSign 2048 bit certificate support
- Generation of ring back after a SIP 183 message followed by a SIP 180 message

SIP 3.2.6 Features (10/11)

- VeriSign intermediate CA certificates
- Updated VeriSign 2048-bit Trusted CA Root Certificate
- RSA 2048 V3 Root certificate
- Event notification version checking configuration parameter
- Full support to RFC 2782

SIP 3.2.5 Features (03/11)

- BLA dialog documents sent within NOTIFY messages ignored
- Enhanced Geo-Redundancy (multiple server fail-over support) feature
- Bridged Line Appearance BLA line dialog rendering converted from no to yes on User Agents that are a remote party to the dialog

SIP 3.2.3 Features (03/10)

Updates

- Sound effects played out of destination based on user configuration

- Retry-after instructions embedded in SIP Response codes 500 and 503 as part of REGISTER and other requests
- Enhancement of appearance on the SoundPoint IP 450 of anti-aliased characters
- Configurable format of DHCP Option 60 Data and additional support for Option 125 as per RFC 3925
- Internal IP address of VVX 1500 no longer being sent in the Facility Message
- Logs do not display Cant set 802.1Q VLAN ID for TCP protocol messages at default (when running on a VLAN)

Added Support

- Support for the SoundStation IP 5000 conference phone
- Network Configuration DHCP sub-menu support for Option 60 format

SIP 3.2.2 Features (12/09)

Updates for All Phones

- iLBC audio code (SoundStation IP 6000 and 7000)
- Formalization of support for DTMF via SIP INFO (initial support in SIP 3.2.0)
- Increased maximum size of contact directory to 128

VVX 1500 Features

- Change of real time operating system
- H.323 signaling protocol support for video
- Webkit browser to replace the XHTML browser
- iLBC audio codec
- H.261 video codec
- Max video bit rate default at 384 kbps
- Curl library version 7.19
- H.235 security
- Mutual TLS authentication
- LLDP protocol
- ITU-T G.719 vocoder
- Dual (SIP/H.323) protocols for outgoing calls
- Video fast update request via RTCP, RFC 5104
- Menu support for H.323
- User accessible menu option to select video call rate.
- Removal of Launchpad Feature

SIP 3.2.1B Features (11/09)

- Support for the SoundPoint IP 335 product

SIP 3.2.0 Features (08/09)

Updates

- Implementation of Scrolling Status bar on SoundPoint IP 320, 321, 330, 331, 550, 560, 650, 650 and SoundStation IP 6000 and 7000
- Microbrowser recognition of multiple mime types
- Replacement of libSRTP algorithms with OpenSSL versions
- DND icon contains text identifying that DND is active

- Addition of ability to take a screen capture
- Addition of Loud Ringer Ringtone selection
- SoundStation IP 7000 Setup Guide
- Requirements met of ETSI TS 102 027-2 v4.1.1 RFC 3261 compliance test for Anatel/Brazil
- Improvement of boot speed in some situations where the boot server is incorrectly configured
- Language selection presentation in appropriate language
- Upgrade of zlib to version 1.2.3
- Upgrade of curl library to version 7.19.2
- Addition of instructions to the SoundPoint IP 450, 550, 560, 650, and 670 for changing label colors in User Guides
- Addition of disabling digit-map rules for Remote Dialing on the SoundStation IP 7000 when connected to an HDX
- Display of idle browser enable/disable from menu
- Navigation button shortcuts in Idle Mode (applies to SoundPoint IP 320, 321, 330, and 331)
- Admin menu option to manually specify the value to be used as the extension displayed on the phone screen (SoundStation IP 7000)
- Improvement Menu items readability when suing Background images on the display
- View status of feature licenses via new menu option
- Removal of Background from scrolling Status bar
- Scrolling Status bar gives equal time to each status message
- Addition of configuration parameters for select ETSI SIP compliance requirements
- Secure entry of passwords in the micro-browser API
- Enable/disable Back soft key in the microbrowser
- Improvement of Enhanced BLF feature when an incoming call occurs whilst viewing BLF monitored line call details
- Inclusion of fmtp attribute specifying Mode=30 in the SDP when 13.33kbps iLBC is used
- Replacement of platform specific TFTP code with tftp support in curl library 7.19.2
- Reduction of local Contact Directory maximum to 99 (SoundPoint IP 430)
- Reduction of maximum number of calls supported to 4 from 8 (SoundPoint IP 430)
- User accessible menu option to display installation of a device certificate
- Easier access to Media Statistics menu
- Configurable behavior for Directed Call Pick-Up as used for Enhanced BLF
- Option to apply digit-map rules to tel:URI initiated calls
- Configuration option for presentation of call appearance on a remotely monitored BLF line
- Configuration option voIpProt.SIP.strictReplacesHeader to control whether the phone requires call-id, to-tag, and from-tag to perform and INVITE with Replaces
- Population of the Display-name field in the To header of responses the phone generates
- Configuration option to send 486 Busy for call rejection
- Combination of SoundPoint IP 550 and 560 User Guides
- Update Destination of outbound call based on the display name in the SIP To header responses
- Call forwarding: user=phone included in refer-to parameter of Refer header
- SDP offer or answer in provisional reliable response and PRACK request and response
- RTP Rx detection and correction for G.722, G.722.1, G.722.1C, and G.719 RTP timestamp increments based on different sample rates
- Equifax Secure eBusiness CA-1 to the trusted CA list
- RFC2543 Hold not working when video SDP present in certain scenarios
- Hook-Flash during POTS calls on the SoundStation IP 7000

Added Support

- iLBC codec on the SoundPoint IP 321, 321, 330, 331, 450, 550, 560, 650, and 670
- Mutual TLS authentication
- LLDP protocol (BootROM 4.2.0 recommended)
- Statically Configured BLF and Call Park and Retrieve enhancements
- Single button Blind Transfer and Retrieve of a call designated as an automata in the dialog used for Statically ConfiguredBLF

- SoundStation IP 7000/HDX6000 Integration (requires future update release to the HDX6000 software)
- Transmission of Join Header as per RFC 3911
- BLF call pick-up using Dialog-info within an INVITE with Replaces header
- iLBC Codec (applies to SoundPoint IP 450, 550, 560, 650, and 670; SoundStation IP 6000 and 7000)

SIP 3.1.8 Features (03/12)

Updates

- Local ringback support on phone when there is a SIP 183 message followed by Sip 180 message
- 2048 bit trusted CA root certificate list

Added Support

- Full support for RFC2782
- VeriSign 2048 bit certificates
- VeriSign Intermediate CA certificates
- RSA 2048 V3 root certificate

SIP 3.1.7 Features (03/11)

- 486 (Busy) response sent to a received INVITE message when call is rejected

SIP 3.1.3C Features (06/09)

- Support for the SoundPoint IP 321 and 331 products

SIP 3.1.3 - Limited Release

Updates

- Corporate Directory: Support for LDAP directory queries using VLV Indexing
- Corporate Directory: Improvements to User Interface
- Corporate Directory: Screen Idle Timeout reset whilst a Corporate Directory search is in process
- Extension of fast-fail over mechanism to transactions initiated over TCP transport
- Network jitter computation and reporting for video packet channels (VVX 1500)
- LCD wake up from the “dim” state to full brightness upon touch (VVX 1500)
- Configuration control of the Dialtone sound level when adding a POTS call to an existing Video call (IP7000/HDX)
- Default for parameter mb.main.idleTimeout changed from 20 to 40 seconds
- Enabling DND/CF Sync prevents the phone from Forwarding or denying any calls that it receives

SIP 3.1.2C Features (06/09)

- Support for the SoundPoint IP 321 and 331 products

SIP 3.1.2B Features (03/09)

- Support for the VVX 1500 product

SIP 3.1.2 Features (01/09) Updates

- Provisioning of license file along with configuration files at application startup
- “Scrolling status bar” on phones to match capability on the SoundPoint IP 450 (applies to all phones except SoundPoint IP 301)

- “Quick Set-Up” option
- Removal of DHCP timeout menu option from UI
- XML API: Softkeys do not allow for having multiple submit buttons on the page containing items list
- Line seized when handset lifted whilst a BLF monitored line is ringing
- BLF indicator for a monitored phone flashes when monitoring phone calls the monitored phone

Added Support

- ACD Call Center Agent functionality using the “Feature Synchronization” method
- Multiple NTP servers via DHCP Options 42 or 4 or DNS SRV or A records

SIP 3.1.1 Features (11/08) Updates

- Revised error message for unsupported USB drives plugged into an IP 650/670 (directs to Polycom support website)
- Improved volume level adjustment capability in hands-free volume control
- “Reset Device Settings” Menu Option clears log files on the phone
- Addition of menu option to enable/disable headset echo cancellation
- SoundPoint IP 450 does not flash Time and Date when time server is not configured

Added Support

- SoundStation IP 7000 integration with HDX Video Systems (Requires BootROM 4.1.2)

SIP 3.1.0C Features (10/08)

- Support for the SoundPoint IP 450 product

SIP 3.1.0 Features (10/08) - Limited Release

Updates

- Re-registration after changing auto parameters
- Robust transfer and call termination behavior against predictable failure modes
- Upload “tech-support” information dump
- Provision of New Call soft key when alerting call appearance is in focus
- EFK: Ability to configure Telephony Soft-Keys
- On-hook dialing during alerting state
- XML API: Micro-browser soft-keys configurable from Server
- Exit will exit, Back will take user back
- Entering “0” and “00” as speed dial number and saving displays invalid Speed Dial number error message
- Warning for entry of duplicate Speed Dial
- Location of Transfer and Conference soft key remains unchanged during Transfer and Conference process
- Configuration to give “dead air” when phone goes off-hook
- Limited number of conference groups to one on SoundStation IP 7000 conference phone
- Updated default list of trusted CAs
- Inclusion of Diversion Header Information in the caller-id display
- Ability to display contents of the SIP warning field to the user
- On register failure (TCPOnly) phone waits 30-60 seconds for retry
- Improved directory configuration parameters

Polycom UC Software Feature Summary

- SAS-VP Provisioning Option hidden from the User Interface
- SIP stack Tx support of Accept-Language
- XML API: Play API – audio file to be downloaded from HTTP server and played using the phone speaker
- CMR/P: Support for USB flash drives larger than 2GB on SoundPoint IP 650/670 phones
- DTMF dialing processes “,” character as 2 second pause
- Deregistration before starting phone reboot process
- EFK: All soft key functions can be mapped to hard keys
- Enhanced BLF: Indication of remote phone ringing to Dialog Package BLF implementation
- Decode support for JPEG image format on SoundStation IP 6000 and 7000 phones
- Diversion Header Information in the caller-id display
- Associate key colors with background bitmaps
- Updated phone UI and Administrator Documents to properly reference “CDP”
- Enhanced BLF: BLF Dialog Handling in SIP Stack
- Enhanced BLF: BLF call appearance UI changes
- EFK: Removal of license requirement from EFK feature
- CMR/P: Increase of recording buffer size
- CMR/P: Rejection of user attempts to perform USB operations while another operation is still in progress
- CMR/P: Addition of UI icon to show when USB drive is busy
- Removal of CFS restriction on SSAWC
- Handset AEC and AES set to “on” in default configuration files
- SoundStation IP 7000: Call lists do not display sip: prefix for URL dialed calls
- Reduction of default maximum memory size for tones from 600 kb to 300 kb

Polycom UC Software Feature Summary

Added Support

- Music On Hold
- Addition of GeoTrust to the built in trusted CA list
- XML API: Support for asynchronous HTTP URL Push and HTTP POST to the micro-browser
- XML API extensions for application support of telephony functions and telephony integration
- Plantronics electronic hook switch (Requires BootROM 4.1.0 or newer)
- EFK: Support for enhanced soft key (ESK) capability
- EFK (Ability to specify a HTTP or HTTPS URL to be loaded by the micro-browser
- Addition of label field to local contact directory
- Addition of ability to invoke internal key functions via the macro engine
- Addition of Slovenian to the list of languages supported by certain SoundPoint/SoundStation IP Phones
- Addition of Polish to the list of languages supported by certain SoundPoint/SoundStation IP Phones
- Addition of configuration to control whether name or number comes first in caller-id
- Addition of EFK support to SoundPoint IP 670

SIP 3.0.4 Features (10/08)

Updates

- Handset AEC and AES set to “on” in default configuration files
- Addition of Speakerphone (Hands Free) volume control

SIP 3.0.3 Features

Updates

- Default boot config changed and packaged sip.cfg value for parameter voice.vad.singalAnnexB
- Addition of Config parameters voIpProt.SIP.strictLineSeize, reg.x.strictLineSeize, and voIpProt.SIP.lineSeize.retries
- SIP stack uses config parameter voIpProt.SIP.strictLineSeize and voIpProt.SIP.lineSeize.retries to make fault-tolerant behavior optional
- Addition of User Option to restart the phone

SIP 3.0.2B Features - Limited Release

Updates

- Dynamic test for un-recognized USB devices
- Call attempt retry on a different line ID after 500 Glare response
- Additional USB flash drives to the internal list of supported drives
- Background preference configuration added to the phone’s configuration web server
- Default LDAP Corporate Directory background re-sync period set to 24 hours
- Initial background LDAP Contact Directory synchronization is optional
- Additional graphic backgrounds to SoundPoint IP 550, 560, and 650 phones

Polycom UC Software Feature Summary

Added Support

- SoundPoint IP 670 phone
- SoundStation IP 6000 conference phone
- SoundStation IP 7000 conference phone
- JPEG images (in addition to BMP format)
- JPEG support to micro-browser

SIP 3.0.1 Features (05/08) – Limited Release

- VLAN Filtering "Off" by default
- Default Corporate Directory background re-sync period to 12 hours

SIP 3.0.0 Features (01/08)
Updates

- RTCP reporting via SIP protocol (except for SPIP 301)
- Statistics gathering and reporting for QOS monitoring according to RFC3611 (except for SoundPoint IP 301)
- Conference Management User Interface for conferences hosted locally on the phone (SoundPoint IP 550, 560, 650)
- Disable speakerphone by configuration file
- "Submit" from Web Browser does not initiate a reconfig/restart when no changes have been made on the phone
- Optional automatic resume on centralized conference
- 4-way conferencing on SoundPoint IP 550, 560, 650 phones
- Integrated with corporate directories using LDAP and Active Directory
- Configurable behavior to support "Single Kepress Conference Set-up"
- Sound effects to accompany USB device insertion and removal
- Call recording and playback on USB flash drive
- SCA Bridging for BroadWorks
- Electronic hook-switch capability using Jabra DHSG protocol (Requires BootROM 4.1.0)
- Handle MIME type application/vq-rtcpxr in SIP stack
- Jabra Jx10 electronic hook switch support (Requires "Interface Cable" from headset base)
- Enhanced speed dial capability
- Full complement of BLF parties on SoundPoint IP 650 plus 3 Ems using UDP
- UI background bitmap configurable on SoundPoint IP 550, 560, and 650 phones
- DHCPINFORM applies if boot server address is null
- DHCPINFORM applies if boot server address is 0.0.0.0
- Reduced DHCPINFORM retry timeouts
- Increased Handset transmit loudness by 3 dB
- Updated XML Dictionaries for Sip 3.0.0

- Lower minimum syslog.renderLevel to 0 from 1

Added Support

- uaCSTA
- Min-expires header

SIP 2.2.2 Features (12/07)

Updates

- De-couple Presence Signaling from Idle Screen Soft-key UI
- Ability to make local contact directory read-only from the phone
- Checking for local contact directory changes during configuration change
- Ability to adjust maximum brightness of SoundPoint IP 550 and 650 phones
- TCP keep-alive on SIP signaling TLS connections
- Ability to adjust maximum brightness of SoundPoint IP Backlit Expansion Modules

Added Support

- SoundPoint 560 product
- SoundPoint IP 320 Part Number 2345-12200-005 and SoundPoint IP 330 Part Number 2345-12200-004 for China market

SIP 2.2.1 Features - Limited Release

- Phone sends TCP Keep-Alive messages to the SIP server when SIP over TLS is configured

SIP 2.2.0 Features (11/07)

Updates

- Return to idle display when no activity occurs in a menu for a configurable period of time
- Sends vendor identifier information through DHCP
- New configurable ring-while-busy options
- General flash file system caching mechanism
- Automatic provisioning support for individual image files
- Synchronization of local DND/CF features with server-based DND/CF features
- Set transfer time-out for image file download to worst case scenario
- Reformat of call list entries
- Configuration option for default transfer type for SoundPoint IP 320 and 300 phones
- Improvement of resistance to denial of service attacks aimed at phone’s web server
- URL dialing terminology change from “Name” to “URL”
- Implementation of 300Hz high pass transmit filter to reduce low frequency noise

- Ability to discover provisioning server address using DHCPINFORM
- Addition of phone serial number (MAC address) to user-agent string HTTP Gets
- Rename of “Services” menu entry to “Applications”
- Addition of low-delay handset acoustic echo canceller for SoundPoint IP 320, 330, 430, 550, and 650 phones
- Phone suppression of DNS queries for 5 minutes (as per RFC 2308 Sec 7.1) if all DNS servers are found unreachable
- Increase of maximum number of registrations on SoundPoint IP 650 to 34
- “Exit” soft key when using microbrowser returns user to telephony application
- Configurable timeout parameter to allow microbrowser to return to telephony application after a period of inactivity in the microbrowser
- Configurable option to display or hide browser status messages in microbrowser
- Boot-up behavior change for idle browser to start about 2 minutes after the phone has booted up to optimize memory use
- Improvement of some translations in Norwegian XML dictionary file
- “Exit” function key on the SoundStation IP 4000 phone when using the microbrowser returns user to telephony application
- Appearance change of soft keys when running microbrowser to look the same when running the telephony application
- Addition of user interface for configuring no-answer and busy forwarding behavior
- Managing TLS custom certificates via the configuration file system
- Specify different versions of the application executable and configuration files in the <Ethernet address>.cfg file on the boot server
- Configuration parameter to control timeout back to the idle display after a period of inactivity in a menu
- Implementation of Ethernet ingress filtering for DoS suppression and VLAN filtering
- Ability to delete the contact number entered in the Forward menu
- Update of all translation dictionary files to rename “Services” menu entry to “Applications”

Added Support

- Microbrowser support for accepting and displaying a URL that points directly to a BMP image
- Microbrowser support for two-dimensional table navigation using all four arrow keys
- Microbrowser support for recognizing mime types
- Tracking of missed calls to be configurable on a per-line basis
- Re-establish a TLS connection if the connection closes
- Support in microbrowser for form functionality when embedded in tbody or out of tbody
- TLS transport Syslog
- Backlit Expansion Module

Polycom UC Software Feature Summary

SIP 2.1.3 Features (03/08)

- Static DNS cache configuration for SIP server Fail-over behavior

SIP 2.1.2 Features (06/07)

- Ability for parameters in <Ethernet address>.cfg to be overridden by model- or platform-specific versions
- Behavior change of the select button or right arrow button in call lists and contact directory on SoundPoint IP 320 and 330
- Addition of configurable failover behavior for authentication signaling
- Addition of configurable option allowing message waiting indicator to be displayed although voicemail cannot be accessed
- Logging of version information for configuration files

SIP 2.1.1C Features (04/07)

- Support for SoundPoint IP 330
- Support for SoundPoint IP 320
- Addition of translations for new phrases needed for SoundPoint IP 320 and 330 phones

SIP 2.1.1 Features (04/07)

- Support for G.729 Annex B SDP signaling per RFC 3555
- Support for 16 levels of gray on the LCD of SoundPoint IP 550 and 650 phones
- Support for new SoundPoint IP 320 and 330 phones in the configuration files

SIP 2.1.0 Features (03/07)

Updates

- Enhanced support for server fall-back configurations
- Microbrowser auto-navigation to first selectable item
- Enhanced “+” global prefix character for E.164 user parts in sip: URIs
- Unique prompt for billing code entry
- Strip or insert leading digits for outgoing calls
- Update of default daylight savings time rules
- Disable message waiting indication on a line by line basis
- Increase of maximum number of digit map segments to 30
- Improvement of text entry efficiency in the microbrowser
- Improved visibility of cursor in text entry fields of microbrowser
- Change of line-seize subscription failure handling to be biased towards providing dial tone
- Addition of more low end dynamic range to volume control

- Set RTP streams to inactive when on hold
- Improvement of “aresDnsLookup: time out on socket select” log message
- Debugging command to display cached DNS NAPTR records
- Call timer clock display changed to have no leading colon
- TCPOnly as a transport option

Added Support

- Table support to microbrowser
- SoundPoint IP 550 platform
- Microbrowser support to the SoundPoint IP 501 platform
- Microbrowser support to the SoundPoint IP 430 platform
- Microbrowser support to the SoundStation IP 4000
- SYSLOG reporting of system status and errors
- Add phone serial number to user agent string in HTTP GET
- Microbrowser support for form input elements with checked = “true” attribute
- Microbrowser support for forms within tables

SIP 2.0.3B Features (12/06)

Updates

- Visual indication of wideband audio

Added Support

- SoundPoint IP 650 platform
- LCD backlight on SoundPoint IP 650
- 32MB of memory on SoundPoint IP 650
- G.722 audio code on SoundPoint IP 650
- 8MB of flash on SoundPoint IP 650
- USB diagnostics

SIP 2.0.2 Features (11/06)

- Split call signaling processing from “lamp management” processing
- Emergency routing is not supported on shared lines

SIP 2.0.1 Features

- Nortel MCP NAT traversal parameters to config files
- Template support in master configuration file

- NAT keep-alive
- Ability to set Ethernet link mode to SoundPoint IP 430
- Ability to set Ethernet link mode to SoundStation IP 4000

SIP 2.0.0 Features (08/06) - Beta Release Only

Updates

- Communications Server 2005
- Communications Server 2005 context
- IM Support with Office Communication and Windows Messenger 5.1 in Microsoft Live
- Addition of Windows Messenger 5.1/Office Communicator-compatible presence and IM
- Addition of IP QoS support for DSCP (DiffServ)
- Option to select specific registration for “presence” signaling
- Useful information logging when phone reboots due to a fatal error
- Configurable SIP re-registration interval
- Caching of the state of the message-waiting indicator LED across controlled reboots
- Increased speed dial menu size limit to 99
- Improved support for multiple m lines in SDP

Added Support

- TLS protocol
- NTLM authentication protocol
- Microsoft Live Communications Server authentication schemes
- Peer-to-peer calls using Microsoft Live Communications Server 2005
- Windows Messenger 5.1 and Office Communicator calling using Microsoft live support in peer-to-peer mode
- Populating speed dial list from a roaming buddies list sent by a Microsoft Live Support for BLF SCA mode
- Platform-specific override strings in dictionaries to allow abbreviated strings for certain platforms
- Multiple redundant provisioning servers
- BroadSoft attendant console/BLF feature
- Individual configuration of secondary dial tone
- Reg.x.address configuration parameter to contain host part

SIP 1.6.7 Features (07/06)

- Ability to set Ethernet link mode on SoundPoint IP 601
- Improved response time of phone to SIP messages
- Configurable line-seize behavior

Polycom UC Software Feature Summary

SIP 1.6.6B Features (05/06)

- Support for SoundPoint IP 430 hardware platform

SIP 1.6.6 Features

- Addition of configurable option to enable phone with BLA to send re-INVITE during conference setup
- Maximum number of buddies increased to 8 (except SoundPoint IP 600 and 601 which can watch 48 buddies)

SIP 1.6.5 Features (02/06)

- Configuration options to allow configuration file parameters to override DHCP values for SNTP server address and GMT offset
- Addition to allow reg.x.address to contain host part instead of being a user part only
- CA certificate expiry no longer checked if SNTP has not been configured
- Addition of flash parameter for SoundPoint IP 601 to toggle power requirements in CDP between 5W and 12W
- Addition of workaround to restart application on the phone if many tasks get unrealistic task delays during startup
- Setting SIP server address from DHCP option 151

SIP 1.6.4 Features (12/05)

- Send and Process “early-only” flag in the “replaces” header to support RFC 3891 in call pickup
- Addition of SP acceptance with telephone-event on the first line
- Disabled CA certificate expiry checking when SNTP has not been configured

SIP 1.6.3 Features (10/05)

- Maximum number of XML retries for SAS-VP to equal to 7 days
- SAS-VP v3 XML configuration transactions
- Setting flash parameters from configuration file
- New dialog event package draft
- New BLA draft

SIP 1.6.1 Features (08/05)

- Directory contact add by pressing and holding an unassigned line key

Polycom UC Software Feature Summary

SIP 1.6.0 Features

Updates

- Addition of Display of Date and Time during a call
- Support for
- Improved Speed Dial key assignment
- Multiple calls from SIP registration (line) can be joined
- Ability to modify forwarding at any time via Forward Menu

Added Support

- SoundPoint IP Expansion Module
- SoundPoint IP 601 hardware platform
- Transfer dispatch during consultation call proceeding state
- Assigned Line Key opens the contact directory

SIP 1.5.3 Features (06/05)

- Upload of Application log file shortly after reboot

SIP 1.5.2 Features (04/05)

- Configuration of Presence and Instant Messaging features disabled by default
- Addition of phone user interface and web interface configuration support for lineKeys and callsPerLineKey

SIP 1.5.1 Features

Updates

- Single Call always shows in the First Call Appearance Position
- Improved Menu Hierarchy
- Conference Feature enhancement to “join” calls in progress
- Flashing time/date until successful SNTP response
- Specify boot server address as URL per RFC 1738
- Application to provision its own configuration files
- Menu entry addition to format the file system
- Display of name and number on incoming caller ID
- Customization options for SSL certificates
- Addition to allow all files in <MAC>.cfg to be full URL’s
- Removal of requirement for at least two audio codecs to be configured
- Merged sip.cfg and ipmid.cfg configuration files into new sip.cfg file
- Disable Time and Date Display
- Split conferences and consultation calls into separate calls

- Allow conference initiation from call hold context
- Default hold signaling changed to RFC 3261 style
- Build ID to software revision stamps in User-Agent header

Added support

- Visual “status” to contacts assigned to Speed Dial Bins
- More than one line key associated with the same SIP identity
- Application support for HTTP and HTTPS boot server transport
- SAS-VP v2 management
  Display useful CID when display name is uninformative
- SoundPoint IP 601 hardware platform
- Arrow-key call-list shortcuts when phone is playing dial tone

SIP 1.4.1 Features (12/04)

- Configurable ‘+’ Global Prefix Character to E.164 User Parts in SIP: URIs

SIP 1.4.0 Features

Updates

- Automatic periodic boot server poll for upgrade
- Phone Restart Menu Command
- Local Conference on SoundStation IP 4000 (When G.729 codec not enabled)
- Integrated Transfer and Conference with Automated Dialing/Speed Dial DHCP VLAN Discovery

Added support

- SoundStation IP 4000 hardware platform
- IP 301 hardware platform

SIP 1.3.1 Features (08/04)

Updates

- ACD Softkeys
- Graphical private label feature
- SIP-based bridged line appearances
- Support for XHTML Microbrowser
- Individual disabling of Call Lists
- Recognition of Instant Message from Windows Messenger 5.0
- Custom call-progress tones to country-specific selection
- New date formats

Added support

- Directed call pick-up
- Group call pick-up
- Call park/retrieve
- True blind transfer
- Centralized conferencing
- Remote missed call notification
- ACD login/logout
- ACD agent available/unavailable
- Server-based last call return
- Basic SIMPLE presence
- SNTP synchronization retry mechanism
- Interoperability support for Windows Messenger 5.0

SIP 1.3.0 Features (06/04)

Basic Features

Missed Call Notification – The phone can display the number of calls missed since the user last looked at the Missed Calls list. The types of calls that are considered as "missed" can be configured per registration. Remote missed-call notification can be used to notify the phone when a call originally destined for it is diverted by another entity such as a SIP server.

Benefits – Keep track of all missed calls.

Bridged Line Appearances - SoundPoint® IP allows calls and lines on multiple phones to be logically related to each other. Mutual exclusion features emulate traditional PBX or key system privacy for shared calls. Incoming calls can be presented to multiple phones simultaneously.
Benefits - A call that is active on one phone is presented visually to phones which share that line.

Idle Display Animation - SoundPoint® IP 500 and IP 600 can display a customized animation on the idle display in addition to the time and date.
Benefits – A quick way to display the status of the user.

Directed Call Pick-Up - SoundPoint® IP allows calls to another phone to be picked up by dialing the extension of the other phone when supported from a SIP server.
Benefits – Instantly communicate with urgent incoming calls from a single workspace.

Group Call Pick-Up - SoundPoint® IP allows calls to another phone within a pre-defined group to be picked up without dialing the extension of the other phone when supported from a SIP server.
Benefits – Provides another way to receive urgent incoming calls from a single workspace.

Call Park/Retrieve - SoundPoint® IP allows active calls to be parked, and the parked call to be retrieved by another phone when supported from a SIP server.

Benefits – Improves management of calls.

Last Call Return - SoundPoint® IP allows server-based last call return when supported from a SIP server.
Benefits – Easily call back and enhance communication productivity.

Advanced Server Features

ACD Login/Logout - SoundPoint® IP allows ACD (Automatic Call Distribution) login and logout when supported from a SIP server.

ACD Agent Available/Unavailable - SoundPoint® IP supports ACD (Automatic Call Distribution) agent available and unavailable when supported from a SIP server.

Accessory Internet Features
MicroBrowser - The home page and proxy can be used by the MicroBrowser when it is selected to provided services and also when it is used as part of the Idle Display. They also control size limits and the Idle Display refresh rate.
Benefits – This provides the content of an idle display which is regularly refreshed.

SIP 1.2.0 Features (02/04)
Basic Features
Message Waiting Indication - SoundPoint® IP flashes a message-waiting indicator LED when instant messages are waiting. The phone can be configured to do so when voice messages are waiting. *Benefits – Keeps track of all calls and messages.*

Shared Call Appearances – A shared line is an address of record managed by a server. The server allows multiple endpoints to register locations against the address of record. SoundPoint® IP allows calls and lines on multiple telephones to be logically related to each other.
Benefits - A call that is active on one telephone is presented visually to telephones which share that appearance.

Localization Features
Customizable Fonts and Indicators – The user interface can be customized by changing the fonts and graphic icons used on the display and the LED indicator patterns. Pre-existing fonts embedded in the software can be overwritten or new fonts can be downloaded, and the bitmaps and bitmap animations used for graphic icons on the display can be changed and repositioned.
Benefits – Enhance user experience and productivity by customizing the phone to the user's familiarity.

Downloadable Fonts – Loaded fonts can either overwrite pre-existing fonts embedded within the software or can extend the phone's font support for Unicode ranges not already embedded.
Benefits - New fonts can be downloaded on to the telephone for greater user experience.

Polycom UC Software Feature Summary

Advanced Server Features
Voicemail Integration - SoundPoint® IP is compatible with voicemail servers. The subscribe contact and call- back mode can be configured per user/registration on the telephone. The phone can be configured with a SIP URL to be called automatically by the telephone.
Benefits – One-touch voicemail access enhances work productivity.

Networking and Security
Incoming Signaling Validation – Three optional levels of security are provided for validating incoming network signaling: Source IP address validation, digest authentication, or both.
Benefits – Allows specification of the type of validation to perform on a request-by-request basis, thereby enhancing protection for secure communication.

Presence and Instant Messaging – Allows monitoring of the status of other users/devices, and through the instant messaging feature the user can send and receive text messages and alerts to incoming messages. This feature can now be disabled via configuration options.

Benefits– Allows full control of the user's work environment.

Emergency/Alternate Proxy or Gateway – Determines the URLS for which they need to be watched, and when one of these defined URLs is detected as having been dialed by the user, the call is automatically be directed to the defined emergency server.

<u>SIP 1.0.9 Features (07/03)</u>
Basic Features
Call log – The phone maintains a log that contains pertinent call information such as remote party identification, time and date, and call duration. The contact information from call log entries can be saved in the contact directory.
Benefits – Saves time and productivity by redialing previous calls.

Call Timer – A call timer is provided on the display. A separate call timer is maintained for each distinct call in progress.
Benefits – Enhances user experience and increase productivity as the end user can view call duration.

Call Waiting – An incoming call during another active call is presented to the user visually on the LCD display.
Benefits - Enhances user experience and efficiency with a visual display of incoming call during an active call.

Called Party Identification – Displays and logs the identity of the remote party specified for outgoing calls. This is the party that the user intends to connect with.
Calling Party Identification – The phone displays the caller identity, derived from the network signaling, when an incoming call is presented.
Benefits – Increases efficiency with a visual display of a previously called party.

Configurable Feature Keys – Phone key functions can be changed from factory defaults.
Benefits - Improves desktop productivity.

Polycom UC Software Feature Summary

Connected Party Identification – The phone displays and logs the identity or the remote party to which the user has connected. The identity is derived from the network signaling.
Benefits – Users can always identify other users with whom they are communicating.

Context Sensitive Volume Control – The volume of user interface sound effects and the receive volume of call audio is adjustable.
Benefits – Establishes an efficient workplace for the user.

Customizable Audio Sound Effects – Sound effects used for incoming call alerts and other indications are customizable. They are composed of patterns of synthesized tones or sample audio files.
Benefits – Users can work in familiar work environments.

Distinctive Incoming Call Treatment – Phones can automatically apply distinctive treatment to calls containing specific attributes. Call attributes that can trigger this include the calling party name or SIP contact (number of or URL format).

Benefits – Predetermines incoming calls for faster response to urgent tasks and communication needs.

Distinctive Ringing - The user can select the ring type for each line. The ring type for specific callers can be assigned in the contact directory. The SIP Alert-Info field can be used to map calls to specific ring types. *Benefits – Enhances user experience as users can aurally identify callers before picking up the phone.*

Distinctive Call Waiting – The SIP Alert-Info field can be used to map calls to distinct call waiting types to two styles, Central and Local.
Benefits – Efficiently manage incoming calls.

Do-Not-Disturb (DND) – Temporarily stops all incoming call alerts. As an option, the phone can be shown as busy while DND is enabled, thereby logging incoming calls as missed for subsequent review.
Benefits – Allows the user to prioritize tasks and control the amount of phone calls received.

Handset, Headset, and Speakerphone – The phone standard includes a handset and a full- duplex speakerphone. A dedicated connector is provided for a headset (not supplied).
Benefits - The phone provides dedicated keys for convenient selection of either the speakerphone or headset, giving the user to choose different options for communication.

Local Contact Directory – The directory can be downloaded from the boot server and subsequently edited locally. Contact information from previous calls may be added for convenient future access. The directory is the central database for several other features including speed-dial, distinctive incoming call treatment, and the presence and instant messaging features.

Benefits – Centralization of various features allows effective management of contact information and phone capabilities.

Local Digit Map – Automate the setup phase of number-only calls.
Benefits *– Eliminates the need for using the Send soft key when making outgoing calls. When a digit pattern matching the digit map is found, the call setup process is completed automatically.*

Polycom UC Software Feature Summary

Microphone Mute - When activated, visual feedback is provided.
Benefits – Increases user communication experience.

Multiple Call Appearances - SoundPoint supports multiple concurrent calls. The hold feature can be used to pause activity on one call and switch to another. When active on one call, an additional incoming call is presented using the familiar call waiting style. Soft keys with call disposition options are presented to the user. The current user interface is limited to two concurrent calls per registration (line); this is an artificial limit and will be expanded in the future.

Benefits – Intuitive interface allows user to effectively manage desktop phone activities.

Soft-key Driven User-Interface – Allows extensive use of intuitive context-sensitive soft-key menus.
Benefits –Familiar interface reduces time needed learning to use the phone and increases work productivity.

Speed-Dial – Entries in the local directory can be linked to the speed-dial system, which allows calls to be placed quickly from dedicated keys as well as from a speed-dial menu.
Benefits – Saves time and increases efficiency for call activities for users.

Time and Date Display – The phone maintains a local clock and calendar. The time and date are displayed in certain operating modes, in particular, when the phone is idle. The clock and calendar must be synchronized to a remote SNTP timeserver.
Benefits – Users can manage their schedules without the need to open up any program on their computer.

Call Management Features

Call Hold – Pause activity on one call to perform another task, such as making or receiving another call. Network signaling is employed to request that the remote party stop sending media and to inform them that they are being held. A configurable local hold reminder feature can be used to remind the user that they have placed calls on hold.
Benefits – Use the desktop phone optimally to meet urgent needs efficiently.

Call Transfer – Enables the user to transform an existing call into a new call between two users. The phone offers both blind transfers and transfers with consultation.
Benefits – Efficient management of calls.

Three-Way Conference, Local – Phones can conference together the local user with the remote parties of two independent calls.
Benefits – Enhances user experience and collaboration.

Call Diversion (Call Forward) – Provides a flexible call diversion feature to divert (forward) calls to another destination. The user’s ability to originate calls is unaffected by all call diversion options. Each registration (line) has its own diversion properties.
Benefits – The call diversion feature works in conjunction with the distinctive incoming call treatment feature.

Automatic Off-hook Call Placement – SoundPoint supports an optional automatic call placement on off-hook feature for each registration.

Audio Processing Features

Low-Delay Audio Packet Transmission – Minimizes latency for audio packet transmission.
Benefits – Communication is uninterrupted or delayed, facilitating greater collaboration and productivity.

Jitter Buffer and Packet Error Concealment - Mitigates packet inter-arrival jitter and out-of- order or lost packets.
Benefits – Communicate in different network environments and minimize resulting negative audio consequences when packets are lost.

Local Conference Mixing – Contains a flexible three-party conferencing capability.
Benefits – Set up local three-party conferences where no external protocol signaling is involved.

Voice Activity Detection (VAD) – Detects periods of relative "silence" in the transmit data path and replaces that silence efficiently with special packets that indicate silence is occurring.
Benefits – Conserve network bandwidth and enhance the focus on the speaking party.

Acoustic Echo Cancellation – Employs advanced acoustic echo cancellation for hands-free operation. Both linear and non-linear techniques are employed to aggressively reduce echo while providing natural full-duplex communication patterns.
Benefits – Facilitate a smooth and natural audio conferencing and reduce listener fatigue.

Background Noise Suppression – Reduces background noise, designed primarily for hands-free Operation.
Benefits – Enhances communication in noisy environments.

Comfort Noise Fill –Employs noise synthesis techniques to smooth out the noise level in the direction toward the remote user.
Benefits – A more natural call experience by providing a consistent noise level to the remote user of a hands- free call.

Automatic Gain Control – Boosts the transmit gain of the local user.
Benefits – Increases the effective user-phone radius and assists with the intelligibility of softer speaking users.

DTMF Tone Generation – Generates and transmits DTMF tones in the RTP streams of connected calls.

DTMF Event RTP Payload – The phone is compatible with RFC 2833 – RTP Payload for DTMF (Dual-Tone Multifrequency) Digits, Telephony Tones, and Telephony Signals. Supported Audio Codecs – G.711µ-law, G.711a-law, G.729AB, SID, RFC 2833.

Polycom UC Software Feature Summary

Presence and Instant Messaging Features
Presence – Allows the phone to monitor the status of other users/devices. Statuses are displayed visually and are in real time. The user can conduct manual or automatic specifications to modify statuses.
Benefits – Increase collaboration productivity by reducing interruptions and communication issue.

Instant Messaging – The user can send and receive instant text messages through the phone, and receives alerts to incoming messages visually and audibly. Messages can be sent from a pre-included list of short messages, or an alphanumeric text entry mode that allows free-form messages using the dial pad.

Benefits – Provides an additional option for users to communicate shorter information instantly.

Localization Features
Multilingual User Interface – The system administrator or the user can choose the language on the phone. *Benefits – Supports major western-European languages and additional languages can be added. Asian languages (Chinese, Japanese, and Korean) render only on the SoundPoint IP 600's higher resolution display.*

Synthesized Call Progress Tones – The phone synthesizes call progress tones during the life cycle of a call, which are easily configurable for compatibility with worldwide telephony standards or local preferences.
Benefit – Emulates the familiar and efficient audible call progress feedback generated by the PSTN and traditional PBX equipment.

Advanced Server Features
Message-Waiting Indication – The phone is compatible with message servers that signal message-waiting indication. The phone can also be configured with a SIP URL to be called automatically by the phone when the user elects to retrieve his/her messages.

Benefits – Allows the user to efficiently organize and retrieve messages.

Multiple Registrations – Each registration is mapped to the familiar concept of a telephone line, and the user can select which line to use for outgoing calls or which registration to use when initiating new instant message dialogues.
Benefits – Facilitates easy management of call activities for users.

Server Redundancy – Phones can be configured with multiple SIP servers, one primary and one or more backup. The phone switches to a backup server when the current primary server fails. Configuration of the backup servers can be static or use advanced DNS methods.

Benefits – Maintained connection to multiple servers prevents communication issues and facilitates productivity.

DNS SIP Server Name Resolution – If a DNS name is given for a proxy/registrar address, the IP address(es) associated with that name are discovered as specified in RFC 3263 (Locating SIP Servers). *Benefit – If a port is given, the only lookup is an A record. If no port is given, NAPTR and SRV records are tried, before falling back on A records if NAPTR and SRV records return no results. If no port is given, and none is found through DNS, 5060 is used.*

Networking and Security - Local user and Administrator Privilege Levels – Local settings menus are protected with two privilege levels, user and administrator, each with its own password. The phone prompts for either the user or administrator password before granting access to the various items in the menus. When the user password is requested, the administrator password also works.
Benefits – Enhanced security for different privilege levels.

SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) – Phones are compatible with the presence and instant messaging features of Windows Messenger and MSN Messenger version 4.6.
Benefits – Take advantage of the Windows Messenger software and improve communication capabilities at the desktop.

Ethernet Switch – The phone contains two Ethernet ports and an embedded Ethernet switch that runs at full line-rate.
Benefits – Allows a PC and other Ethernet devices to connect to the office LAN by daisy chaining through the phone, eliminating the need for a standalone hub.

RTP Ports – Compatible with RFC 1889 – RTP: A Transport Protocol for Real-Time Applications.
The phone treats all RTP streams as bi-directional from a control perspective and expects that both RTP endpoints will negotiate the respective destination IP addresses and ports.
Benefits – Allows RTCP to operate correctly even when RTP media may flow in only one or no direction, and allows greater security-packets from unauthorized sources to be rejected.

Upgrading and Rebooting – Both the bootROM and the application executable can be updated automatically via the centralized provisioning (boot server) model.
Benefits – Keep up with the most recent updates for maximum productivity out of the phone.

Event Logging – Phones maintain boot and application event log files. They are stored in the phone's flash file system and are periodically uploaded to the phone's home directory on the provisioning boot server if permitted by security policy.
Benefits – Use stored files to identify and diagnose problems.

Working with Network Address Translation (NAT) – Phone signaling and RTP traffic use symmetric ports and the external IP address and ports used by the NAT on the phone's behalf can be configured on a per-phone basis.

Audio Quality Issues and VLANs – Phones contain both network layer and Ethernet layer support for prioritizing voice and signaling traffic over the network. QoS parameters include IP TOS (Type-of- service) bits, and Ethernet IEEE 802.1p user priority. Phones also support RTCP per RFC 1889.

Transfer – Phones support the transfer protocol utilizing the REFER method specified in draft- ietf-sip-cc-transfer-05, but also support the older transfer method utilizing Bye-also.